Patent claims

1.   A method for improving the quality of an audio
transmission in which audio data containing samples
5   (A1, ..., A8) of an audio signal are asynchronously
transmitted in data packets from a transmitting
communication system (PBX1) via a packet-oriented
communication network (LAN) to a receiving
communication system (PBX2) and

10   an information item relating to the transmission of
data packets is detected, characterized in that the
audio data are converted in the sense of an alteration
of their sampling rate by means of digital filtering,
the sampling rate being altered in dependence on the
15   detected information item, in such a manner that due to
the altered sampling rate, a quality of service of the
audio transmission is optimized with regard to a
current transmission situation indicated by the
detected information item.

20   2.   The method for improving the quality of an
audio transmission in which audio data containing
samples of an audio signal are asynchronously
transmitted in data packets from a transmitting
communication system (PBX1) via a packet-oriented
25   communication network (LAN) to a receiving
communication system (PBX2) and

an information item relating to the transmission of
data packets is detected, characterized in that the
audio data are digitally converted in the sense of a
30   modification of the duration (L1+2*L2+L3) of an audio
signal represented by the audio data whilst largely
retaining a pitch of the audio signal, the duration
(L1+2*L2+L3) being modified in dependence on the
detected information item, in such a manner that due to
35   the modified duration (L1+L2+L3), a quality of service
of the audio transmission is optimized with regard to a
current transmission situation indicated by the
detected information item.

3.   The method as claimed in claims 1 and 2.

4.   The method as claimed in one of the preceding claims, characterized in that the audio data to be transmitted are converted by the transmitting

5   communication system (PBX1) and a conversion message about the conversion is transmitted from the transmitting communication system to the receiving communication system (PBX2).

5.   The method as claimed in claim 4, characterized

10 in that the transmitted audio data are reconverted by the receiving communication system (PBX2), the change in the audio data taking place in the reconversion being determined by means of the conversion message transmitted.

15 6.   The method as claimed in one of the preceding claims, characterized in that the transmission of the data packets is monitored by the receiving communication system (PBX2) and an information item relating to this transmission is transmitted to the

20 transmitting communication system (PBX1) and that the audio data are converted by the transmitting communication system (PBX1) in dependence on the information item transmitted.

7.   The method as claimed in claim 6, characterized

25 in that the information item transmitted specifies a data packet loss rate and that, if the data packet loss rate rises, the audio data are converted by the transmitting communication system (PBX1) in such a manner that the audio data rate is reduced.

30 8.   The method as claimed in one of the preceding claims, characterized in that a detected incorrect adaptation of the data rate of the received audio data is at least partially compensated by the receiving communication system (PBX2) by means of a conversion of

35 the received audio data.

9.   The method as claimed in one of the preceding claims, characterized in that the received audio data are converted after having been read out of an input

buffer provided for compensating data packet delay variations, in which the read-out speed of the input buffer is controlled by a change in the audio data rate due to the conversion.

5   10.   The method as claimed in one of the preceding claims, characterized in that, in the case of the loss of a data packet, the audio data contained in a data packet preceding and/or following the lost data packet are converted by the receiving communication system

10   (PBX2) in the sense of an extension of the duration of an audio signal represented by the audio data, in such a manner that the extension of the duration at least shortens a gap in the audio signal due to the lost data packet.

15   11.   A communication system (PBX1, PBX2) for transmitting and/or receiving audio data containing samples (A1, ..., A8) of an audio signal via a packet-oriented communication network (LAN), comprising

      a monitoring means (W1, W2) for detecting an

20   information item relating to the transmission of data packets containing audio data,
      characterized by a digital sampling rate conversion device (AU1A, AU1B, AU2A, AU2B) for converting the audio data in the sense of altering their sampling rate

25   and
      a control means (ST1, ST2) for controlling the sampling rate alteration in dependence on the information item detected.

      12.   The communication system (PBX1, PBX2) for

30   transmitting and/or receiving audio data containing samples of an audio signal via a packet-oriented communication network (LAN), comprising
      a monitoring means (W1, W2) for detecting an information item relating to the transmission of data

35   packets containing audio data,
      characterized by a digital timescale conversion device (ZU1A, ZU1B, ZU2A, ZU2B) for converting the audio data in the sense of changing the duration (L1+2*L2+L3) of

an audio signal represented by the audio data whilst largely retaining a pitch of the audio signal, and

a control means (ST1, ST2) for controlling the change in duration in dependence on the information item

5 detected.

13.   The communication system as claimed in claim 11 and 12.

14.   The communication system as claimed in claim 11 or 13, characterized in that the sampling rate

10 conversion device (AU1A, AU1B, AU2A, AU2B) exhibits a digital filter chip for converting the audio data.

15.   The communication system as claimed in claim 12 or 13, characterized in that the timescale conversion device (L1+2*L2+L3) exhibits a digital signal processor

15 for converting the audio data.